fw

UNITED STATES PATENT AND TRADEMARK OFFICE

UNITED STATES DEPARTMENT OF COMMERCE
United States Patent and Trademark Office
Address: COMMISSIONER FOR PATENTS
P.O. Box 1450
Alexandria, Virginia 22313-1450
www.uspto.gov

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/700,488	11/05/2003	Kiryung Lee	123056-05004473	6286

43569 7590 05/07/2007
MAYER, BROWN, ROWE & MAW LLP
1909 K STREET, N.W.
WASHINGTON, DC 20006

EXAMINER	
SIEDLER, DOROTHY S	

ART UNIT	PAPER NUMBER
2626	

MAIL DATE	DELIVERY MODE
05/07/2007	PAPER

Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)	
	10/700,488	LEE ET AL.	
	Examiner	Art Unit	
	Dorothy Sarah Siedler	2626	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *05 November 2003*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-7* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-7* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *11-5-03* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☒ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☒ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date *11-5-03*.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

This is the initial response to the application filed November 5, 2003. Claims 1-7 are pending and are considered below.

Claim Objections

Claims 2, 6 and 7 are objected to because of the following informalities:

Claim 2 recites "WNR", however a specific definition of the acronym is not provided. In addition, claims 2 uses variables, such as α,x_n and s_n, which are not defined.

Claims 6 and 7 use the variable Δ_e which is not specifically defined.

Appropriate correction is required.

Claim Rejections - 35 USC § 112

The following is a quotation of the second paragraph of 35 U.S.C. 112:

> The specification shall conclude with one or more claims particularly pointing out and distinctly claiming the subject matter which the applicant regards as his invention.

Claims 2,3 and 6 are rejected under 35 U.S.C. 112, second paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter which applicant regards as the invention.

Claim 2 recites the terms "host signal" and "audio signal" which are both used to refer to the variable x_n, therefore making in unclear which signal x_n corresponds to. The examiner considers the use of " an audio signal" as a typographical error, and interprets

it as " an host signal", this interpretation used throughout the remainder of this application.

Claim 2 also recites the limitation "the scale α ". There is insufficient antecedent basis for this limitation in the claim. The examiner interprets "the scale α " as the scale factor commonly used during uniform scalar quantization, as is know in the art. This interpretation used throughout the remainder of this office action.

Claims 3 and 6 recite, "estimating a scale factor", however it is unclear by this term if the scale factor refers to the scale factor used during uniform scalar coding at the encoder, or a value of the amplitude modification, or scaling, of the watermarked signal as it is transmitted and received at the decoder. The examiner interprets "estimating a scale factor" as "estimating an amplitude scale rate", i.e. estimating the amplitude modification of the encoded watermarked signal received at the decoder. This interpretation is used throughout the remainder of this office action.

Claim 3 recites the limitation "the amplitude-scaling". There is insufficient antecedent basis for this limitation in the claim. Therefore the examiner interprets this as "the amplitude scale rate", this interpretation used throughout the remainder of this office action.

Claim Rejections - 35 USC § 103

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

Claims 1,4 and 5 are rejected under 35 U.S.C. 103(a) as being unpatentable over ***Tewfik*** (6,061,793) in view of ***Sudharsanan*** (5,764,698).

As per claims 1 and 4, ***Tewfik*** discloses an amplitude-scaling resilient audio watermarking encoding apparatus based on a quantization, comprising: a filterbank for dividing an inputted audio signal into a plurality of subbands (column 4 lines 37-38); a psychoacoustic module for applying a psychoacoustic model to the inputted audio signal to provide a signal-to-mask ratio (SMR) (column 4 lines 33-35 and 38-40); a synthesis filterbank for synthesizing the divided and watermarked subband signals to output a watermarked audio signal (Figure 4(b), *the output of the encoder is an audio signal comprising the original signal plus the watermark, therefore it is inherent that the subbands were synthesized in order to output an audio signal*), and transmitting watermarked audio signal and the encoding parameter (column 1 lines, *watermarking is used to provide proof of authorship in media data, therefore the watermarked signal and the encoding parameter must be sent, in order to validate the proof of ownership*).

Tewfik does not explicitly disclose the use of a polyphase filterbank to divide the audio signal into subbands, or a watermark encoder for evaluating an encoding parameter from the plurality of subbands according to the signal-to-mask ratio (SMR) provided from the psychoacoustic module and embedding the encoding parameter and a watermark into subbands having middle frequency subbands among the plurality of subbands. However, ***Tewfik*** does disclose that in order to provide for robustness of data, hidden data is often spread out amongst a spectrum, including middle frequency subbands, using known spread-spectrum techniques (column 3 lines 28-30). In addition, ***Sudharsanan*** discloses that, according the MPEG audio encoding standard, an audio signal is divided into subbands (column 3 lines 48-49), and the signal-to-mask ratio for each subband is determined, then used to determine the quantizer step value (encoding parameter)(column 4 lines 64-67); the scale factor and the quantizer step size then sent to a receiver for use during decoding. Also, official notice is taken that it is well known on the art to use a polyphase filterbank for dividing an input audio signal into a plurality of subbands. Polyphase quadrature filters are used as part of MPEG-1 layer I and II, in order to provide a subband-coding scheme which attempts to preserve the original signal as much as possible using bit allocation based on the psychoacoustic model.

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to have a watermark encoder for evaluating an encoding parameter from the plurality of subbands according to the signal-to-mask ratio (SMR) provided from the psychoacoustic module and embedding the encoding parameter and a watermark into subbands having middle frequency subbands among the plurality of

subbands in ***Tewfik***, since the SMR is a known and reliable method of determining a masking threshold; also, embedding parameters at middle frequency subbands enables robustness of data, as indicated in ***Tewfik*** (column 3 lines 30-35).

In addition, it would have been obvious to one of ordinary skill in the art at the time of the invention to use a polyphase filterbank in ***Tewfik***, since it is a know method for producing subbands from an audio signal, with available software or hardware implementation techniques.

As per claim 5, ***Tewfik*** in view of ***Sudharsanan*** disclose the method of claim 4, and ***Tewfik*** further discloses wherein the step of encoding the watermark is performed by embedding the watermark in middle frequency subbands (column 3 lines 28-30, *hidden data is spread amongst the frequency spectrum, including middle frequency subbands, based on known spread-spectrum techniques*).

Claim 2 is rejected under 35 U.S.C. 103(a) as being unpatentable over ***Tewfik*** in view of ***Sudharsanan***, and further in view of ***Eggers*** ("Estimation of Amplitude Modifications before SCS Watermark Detection" SPIE January 2002).

As per claim 2, ***Tewfik*** in view of ***Sudharsanan*** disclose the amplitude-scaling resilient audio watermarking encoding apparatus of claim 1, and ***Sudharsanan*** further discloses wherein the watermark encoder includes: a parameter evaluator for evaluating the

encoding parameter value (.DELTA..sub.e,.alpha.) from the signal-to-mask ratio provided from the psychoacoustic model (column 4 lines 45 – 67, *a scale factor is determined, as well as a quantizer step size using the SMR*). However neither ***Tewfik*** or ***Sudharsanan*** disclose an estimation value (WNR) of a noise intensity determined by a specification of a lossy compression; a quantizer for performing an uniform scalar quantization with respect to an audio signal x.sub.n according to the quantizer step size .DELTA..sub.e of an encoder by using a quantizer selected by a watermark d.sub.n; an adder for subtracting the host signal x.sub.n from an output of the quantizer; a multiplier for multiplying an output of the adder by the scale .alpha.; and an adder for adding an output of the multiplier to the host signal x.sub.n to output a watermarked subband signal s.sub.n. ***Eggers*** discloses an estimation value (WNR) of a noise intensity (page 8, first paragraph, *an estimation of system performance is calculated using for a range of WNR values (estimation WNR value)*), a quantizer for performing uniform scalar quantization according to quantizer step size selected by a watermark (page 2, section 2. SCS Watermarking and AWGN and Amplitude Scaling Attack, third paragraph, *the system is quantized using scalar uniform quantizers, where the quantizer step size is codebook parameter*), an adder for subtracting the host signal x.sub.n from an output of the quantizer, a multiplier for multiplying an output of the adder by the scale .alpha.,and an adder for adding an output of the multiplier to the host signal x.sub.n to output a watermarked subband signal s.sub.n (page 2, equation (3)).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to use an estimation value (WNR), perform uniform scalar quantization

according to a quantizer step size, have an adder for subtracting the host signal x.sub.n from an output of the quantizer, a multiplier for multiplying an output of the adder by the scale .alpha , and an adder for adding an output of the multiplier to the host signal x.sub.n to output a watermarked subband signal s.sub.n in ***Tewfik*** and ***Sudharsanan***, since the WNR value can be used to estimate the accuracy of different pilot sequences, as indicated in ***Eggers*** (page 7-8, section 3.4 Estimation performance for different L_{pilot}). In addition, uniform scalar quantization and the aforementioned mathematical operations on the host signal provide a watermarking scheme, based on the scalar costa scheme, which provides more reliable communication, even for strong attacks, as indicated in ***Eggers*** (page 1, section 1.Introduction).

Claims 3,6 and 7 are rejected under 35 U.S.C. 103(a) as being unpatentable over ***Eggers*** in view of ***Moon*** ("The Expectation-Maximization Algorithmn" IEEE 1996), further in view of ***Tewfik***, and further in view of ***Quatieri*** (Discrete –Time Speech Signal Processing, Principles and Practice, Prentice Hall 2002)

As per claim 3, ***Eggers*** discloses an amplitude-scaling resilient audio watermarking decoding apparatus based on a quantization, comprising: estimating an scale factor (amplitude scale rate) from an encoding parameter contained in the received audio signal and a watermarked subband (page 1, section 1.Introduction, third and fourth paragraphs, *the system is designed to estimate amplitude modifications prior to Scalar Costa Scheme watermark detection*), a watermark decoder for extracting a watermark

considering the quantizer step size (page 3, section 3. Estimation based on SCS pilot sequences, *the adjusted quantization step size is used for watermark reception*), and generating the quantizer step size .DELTA..sub.d of a decoder according to the amplitude-scaling (page 3, section 3. Estimation based on SCS pilot sequences, first paragraph, *the quantizer step size used at the receiver, or decoder, is adjusted based on the scale factor g*). However, ***Eggers*** does not explicitly state a polyphase filterbank for dividing a received audio signal into the predetermined number of subbands, an expectation maximization (EM) estimator for estimating an scale factor from an encoding parameter contained in the received audio signal and a watermarked subband according to an EM algorithm, a watermark decoder for extracting a watermark from a subband corresponding to the middle frequency considering the quantizer step size, and an integrated determiner for integrating outputs of the watermark decoder to determine a watermark. Official notice is taken that it is well known on the art to use a polyphase filterbank for dividing an input audio signal into a plurality of subbands. Polyphase quadrature filters are used as part of MPEG-1 layer I and II, in order to provide a subband-coding scheme which attempts to preserve the original signal as much as possible using bit allocation based on the psychoacoustic model. In addition, ***Quatieri*** discloses that subband coding is one common type of encoding for audio signals, in which a filterbank is used to separate an audio signal into frequency bands, or subbands, including middle frequency bands (page 621-625, section 12.5.1). ***Tewfik*** also discloses a system where a watermark, or hidden data, is spread against a signature into bands including the middle frequency bands, according to known spread

spectrum techniques (column 3 lines 28-30). Therefore a system using subband coding must extract a watermark from a subband corresponding to a middle frequency. ***Quatieri*** also discloses that subband coding is based on the general filter-bank summation method (page 621, section 12.5.1), which separates a signal into frequency bands, then uses a synthesis filter (page 324, equation 7.14) to integrate each sequence to determine the original signal. In addition, without direct access to data necessary to estimate the parameters, In addition, ***Moon*** discloses that the EM algorithm is commonly used for signal processing applications where parameters are estimated without direct access to data necessary to estimate the parameters, or when data is missing (page 47, first paragraph).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to use subband coding, and a polyphase filterbank, to separate an incoming signal into subbands, extract a watermark from the subband corresponding to the middle frequency, then integrate the outputs of the watermark decoder to determine the watermark in ***Eggers***, since subband coding, using a polyphase filterbank, is a know method for producing subbands from an audio signal, with available software or hardware implementation techniques. In addition, subband coding limits quantization noise to the band it was generated from, and can cancel out quantization noise that leaks across bands, with the use of overlapping filters, as indicated in ***Quatieri*** (page 621-622).

It would also have been obvious to one of ordinary skill in the art at the time of the invention to use the EM algorithm to estimate a scale factor (amplitude scale rate)

from an encoding parameter contained in the received audio signal and a watermarked subband in ***Eggers***, since the EM algorithm is a known and reliable algorithm for the estimation of signal parameters without direct access to data necessary to estimate the parameters, as indicated in ***Moon*** (page 47, first paragraph), with available software or hardware implementation techniques.

As per claim 6, ***Eggers*** discloses a method for decoding an audio signal comprising the steps of: receiving the audio signal and a side information (page 1, section 1. Introduction, *the scalar costa scheme is used which includes using side information, or a pilot sequence, embedded with the watermark*); estimating an scale factor from the side information and the received audio signal, and evaluating the quantizer step size of a decoder from the estimated amplitude-scale rate (page 3, section 3. Estimation based on SCS pilot sequences, *when the encoding quantizer step size is known at the receiver, the scale factor g can be derived, which is then used to adjust the quantizer step size at the receiver*); decoding a watermark considering the evaluated quantizer step size (page 3. Estimation based on SCS pilot sequences, first paragraph, *the adjusted quantizer step size is sufficient t enable watermark reception*). However ***Eggers*** does not disclose dividing the audio signal into subbands, estimating an scale factor from the side information and the received audio signal by using an expectation maximization (EM) algorithm, decoding a watermark from the subbands considering the evaluated quantizer step size, and summing up the decoded values to calculate an average, and calculating a correlation between the average and codes of a codebook to

obtain a watermark. ***Quatieri*** discloses that subband coding is one common type of encoding for audio signals, in which a filterbank is used to separate an audio signal into frequency bands, or subbands, including middle frequency bands (page 621-625, section 12.5.1). ***Tewfik*** also discloses a system where a watermark, or hidden data, is spread against a signature into bands including the middle frequency bands, according to known spread spectrum techniques (column 3 lines 28-30). Therefore a system using subband coding must extract a watermark from a subband corresponding to a middle frequency. ***Quatieri*** also discloses that subband coding is based on the general filter-bank summation method (page 621, section 12.5.1), which separates a signal into frequency bands, then uses a synthesis filter (page 324, equation 7.16, *the discrete version of equation 7.14*) to sum up each sequence and calculate an average to determine the original signal. The step of calculating a correlation between the average codes and codes of a codebook is simply looking up the codewords in a codebook, which ***Quatieri*** also discloses as a known step to scalar quantization (pages 598-600, section 12.3 Scalar Quantization). In addition, ***Moon*** discloses that the EM algorithm is commonly used for signal processing applications where parameters are estimated without direct access to data necessary to estimate the parameters, or when data is missing (page 47, first paragraph).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to use subband coding to separate an incoming signal into subbands, decode a watermark from the subbands considering the quantizer step size, sum the decoded values to calculate an average, then calculate a correlation between the

average codes and codes in a codebook in ***Eggers***, since subband coding is a know method for producing subbands from an audio signal, with available software or hardware implementation techniques. In addition, subband coding limits quantization noise to the band it was generated from, and can cancel out quantization noise that leaks across bands, with the use of overlapping filters, as indicated in ***Quatieri*** (page 621-622).

It would also have been obvious to one of ordinary skill in the art at the time of the invention to use the EM algorithm to estimate a scale factor (amplitude scale rate) from an encoding parameter contained in the received audio signal and a watermarked subband in ***Eggers***, since the EM algorithm is a known and reliable algorithm for the estimation of signal parameters without direct access to data necessary to estimate the parameters, as indicated in ***Moon*** (page 47, first paragraph), with available software or hardware implementation techniques.

As per claim 7, ***Eggers*** discloses wherein the quantizer step size .DELTA..sub.d is calculated by multiplying the received quantizer step size of the encoder by the estimated scale factor (page 3, section 3. Estimation based on SCS pilot sequences, *the estimation of the received quantizer set size is based on the encoding quantizer step size multiplied by the scale factor*).

Conclusion

The prior art made of record and not relied upon is considered pertinent to applicant's disclosure. Please see PTO-892 form.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Dorothy Sarah Siedler whose telephone number is 571-270-1067. The examiner can normally be reached on Mon-Thur 9:30am-5:30pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on 571-272-7602. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

DSS

TĀLIVALDIS IVARS ŠMITS
PRIMARY EXAMINER